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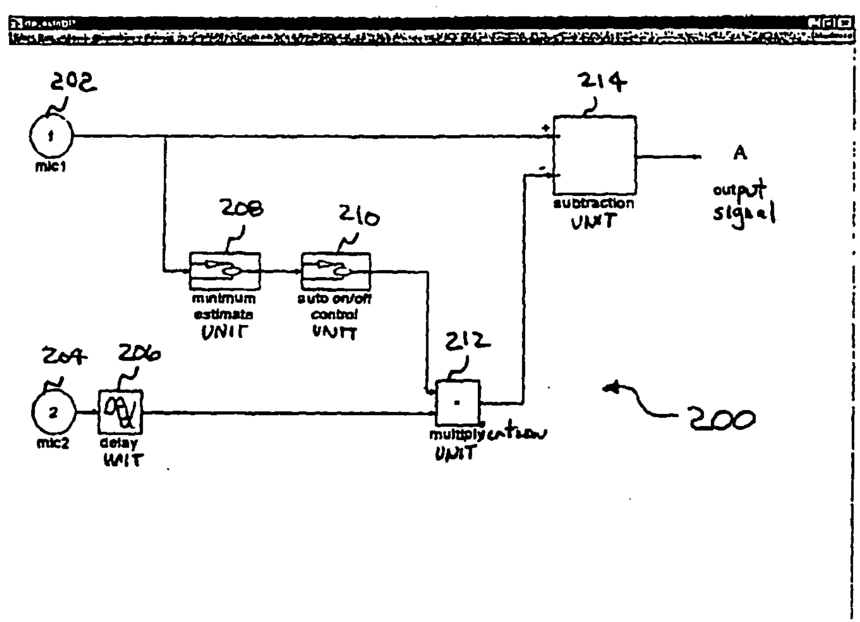

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(57) **Abstract:** Improved approaches for directional processing in multi-microphone processing systems are disclosed. According to one aspect, these approaches operate to control activation/deactivation of directional processing in multi-microphone processing systems. According to another aspect, these improved approaches can adaptively suppress interfering noise in a multi-microphone directional system. These approaches are particularly useful for hearing aid applications in which directional noise suppression is important.

For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.

DIRECTIONAL PROCESSING FOR MULTI-MICROPHONE SYSTEM

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates to multi-microphone sound pick-up systems and, more particularly, to directional processing in multi-microphone sound pick-up systems.

2. Description of the Related Art

Suppressing interfering noise is still a major challenge for most communication devices involving a sound pick up system such as a microphone or a multi-microphone array. The multi-microphone array can selectively enhance sounds coming from certain directions while suppressing interference coming from other directions.

FIG.1 shows a typical directional processing system in a two-microphone hearing aid. The two microphones pick-up sounds and convert them into electronic or digital signals. The output signal from the second microphone is delayed and subtracted from the output signal of the first microphone. The result is a signal with interference from certain directions being suppressed. In other words, the output signal is dependent on which directions the input signals come from. Therefore, the system is directional. The physical distance between the two microphones and the delay are two variables that control the characteristics of the directionality. For hearing aid applications, the physical distance is limited by the physical dimension of the hearing aid. The delay can be set in a delta-sigma analog-to-digital converter (A/D) or by use of an all-pass filter.

Although the typical directional processing system, such as shown in FIG. 1, is able to suppress interference from certain directions, the typical directional processing also has some disadvantages. One disadvantage is that the frequency response of the typical directional processing system is like a

high-pass filter, with low frequency components attenuated more than high frequency components. This is so-called a low frequency roll-off phenomenon. Another disadvantage is that the noise floor of the typical directional processing system is higher than with a single microphone. This is because each microphone has a noise floor. The typical directional processing system has more than one microphone and the combined noise floor of two microphones is always higher than that of a single microphone. Accordingly, it is desirable to turn-off the directional processing during quiet periods to avoid these two disadvantages.

Most existing hearing aids that perform directional processing provide a manual means for users to turn the directional processing on or off. Recently, U.S. Patent 5,214,709 proposed a method to turn the directional processing on/off simply based on the level of the microphone response. One problem with such a design is that the turning of the directional processing on/off is not based on whether the environment is noisy or quiet. As a result, high-level clean speech could trigger the directional processing even though unwanted. Further, because the triggering of directional processing is simply based on a voltage level of the microphone response, the fluctuation in speech signal could undesirably turn the directional processing on and off, which is very annoying for users.

Thus, there is a need for improved approaches to control directional processing in multi-microphone processing systems.

As noted previously, a multi-microphone array can selectively enhance sounds coming from certain directions while suppressing interferes coming from other directions. The pattern of the direction selection can be fixed or adaptive. Adaptive selection is more attractive because it intends to maximize SNR depending on the sound environment. However, because the relative low frequency range of audio applications, existing adaptation techniques are effective only for microphone array with large physical dimension. For applications where physical dimension is limited, such as the case in hearing aid applications, traditional adaptation using Finite-Impulse-Response (FIR) adaptive filtering techniques is not effective. As a result, most hearing aids that have directional processing can only give a fixed directional pattern which is

effective in improving Signal-to-Noise Ratio (SNR) in some conditions but less effective in other conditions.

FIGs. 9(a) - 9(c) illustrate polar patterns of a directional processing system corresponding to three different delay values. The term "polar pattern" has often been used to describe the characteristics of a directional processing system. The physical distance between the two microphones of the directional processing system is fixed. When a sound source is at 0 degrees, which is the direction along the axis of the two microphones and on the side of the front microphone, the directional processing system has a maximum output. When the sound source is away from 0 degrees, the output is reduced. The direction at which the output of the directional processing system has a maximum reduction is called directional null. Ideally, the directional null occurs at the direction of an unwanted noise source. The location of the directional null is related to the value of the delay. If the noise source is in the direction of 180 degrees, the delay should be set to a value so that the polar pattern is a cardioids with the directional null at 180 degrees (see FIG. 9(a)). If the noise source is in the direction of 115 degrees, the delay should be set to a value so that the polar pattern is a hyper-cardioid with the directional null at 115 degrees (see FIG. 9(b)). If the noise source is in the direction of 90 degrees, the delay should be set to a value so that the polar pattern is a bi-directional with the directional null at 90 degrees (see FIG. 9(c)). Ideally, the delay should be set in such a way that the null is placed in the direction of the dominant noise source so that the noise can be highly suppressed. If the direction of the noise source is known, the optima delay can be calculated as:

$$delay = d/c*\cos(180^0 - q),$$

where d is distance of the two microphones, c is sound propagation speed, and q is direction angle in degree of the noise source.

One problem with conventional noise suppression approaches is that the direction of a noise source to be suppressed by the directional processing is often unknown. Conventionally, the estimating of the direction of a noise source is difficult because the frequency of audio sounds is relative low. The direction of the noise source is often merely a rough estimate from which a

delay is then fixed to provide directional processing. In fact, most hearing aids currently available in the market merely set the delay to a fixed value so that directional processing has a fixed polar pattern for all conditions. Unfortunately, the noise suppression of such devices is often inadequate because the noise source is often at a direction other than that corresponding to the fixed delay.

Thus, there is also a need for improved approaches to directional processing by adapting a directional null according to the direction of interfering noise source.

SUMMARY OF THE INVENTION

Broadly speaking, the invention relates to improved approaches adaptively suppress interfering noise in a multi-microphone directional system. These approaches are particularly useful for hearing aid applications in which directional noise suppression is important.

According to one aspect of the invention, the approaches control directional processing in multi-microphone processing systems. These approaches operate to control activation/deactivation of directional processing in multi-microphone processing systems. As a result, directional processing can be automatically activated or deactivated based on the amount of interference (e.g., noise) in a listening environment.

According to another aspect of the invention, the approaches adjust a delay adaptively so that a directional null is placed in the direction of a dominant noise source. This would produce maximum Signal-to-Noise Ratio (SNR) improvement across all conditions. In other words, the dominant noise source is attenuated (e.g., suppressed) but the desired sound from a particular direction is not attenuated.

The invention can be implemented in numerous ways including as a method, system, apparatus, device, and computer readable medium. Several embodiments of the invention are discussed below.

Other aspects and advantages of the invention will become apparent from the following detailed description taken in conjunction with the accompanying drawings which illustrate, by way of example, the principles of the invention.

BRIEF DESCRIPTION OF THE DRAWINGS

The invention will be readily understood by the following detailed description in conjunction with the accompanying drawings, wherein like reference numerals designate like structural elements, and in which:

FIG.1 shows a typical direction processing system in a two-microphone hearing aid;

FIG. 2 is a block diagram of a two-microphone directional processing system according to one embodiment of the invention;

FIG. 3 is a block diagram of a minimum estimate unit according to one embodiment of the invention;

FIG. 4 is a block diagram of a minimum estimate unit according to another embodiment of the invention;

FIG. 5 is a block diagram of an automatic on/off control unit according to one embodiment of the invention;

FIG. 6 is a schematic diagram of an automatic on/off control unit according to one embodiment of the invention;

FIG. 7 is a graph illustrating a relationship between directional processing (indicated by directional scale) and an input level for the automatic on/off control unit illustrated in FIG. 5;

FIG. 8 is a graph illustrating a relationship between directional processing (indicated by directional scale) and an input level for the automatic on/off control unit illustrated in FIG. 6;

FIGs. 9(a) - 9(c) illustrate polar patterns of a directional processing system corresponding to three different delay values;

FIG. 10 is a block diagram of a two-microphone directional processing system according to one embodiment of the invention;

FIG. 11 shows a block diagram of an optimal delay determination unit according to one embodiment of the invention;

FIG. 12A is a block diagram of a delay generator according to one embodiment of the invention;

FIG. 12B is a schematic diagram of a circuit suitable for use as a delay increment calculation circuit according to one embodiment of the invention;

FIG. 12C is a schematic diagram of a circuit suitable for use as a delay increment calculation circuit according to another embodiment of the invention;

FIG. 12D is a schematic diagram of a circuit suitable for use as the delay increment calculation circuit according to still another embodiment of the invention;

FIG. 13 shows an alternative method for adapting the direction null to maximize SNR in a two-microphone directional processing system;

FIG. 14 is a graph illustrating a spectrum of a 1 kHz pure tone in white noise without any directional processing for noise reduction;

FIG. 15 is a graph illustrating a spectrum of a 1 kHz pure tone in white noise with fixed-pattern (hypercaidiod) directional processing for noise reduction; and

FIG. 16 is a graph illustrating a spectrum of a 1 kHz pure tone in white noise with adaptive directional processing according to one embodiment of the invention to provide enhanced noise reduction.

DETAILED DESCRIPTION OF THE INVENTION

The invention relates to improved approaches to adaptively suppress interfering noise in a multi-microphone directional system. These approaches are particularly useful for hearing aid applications in which directional noise suppression is important.

According to a first aspect of the invention, the approaches control directional processing in multi-microphone processing systems. These approaches operate to control activation/deactivation of directional processing in multi-microphone processing systems. As a result, directional processing can be automatically activated or deactivated based on the amount of interference (e.g., noise) in a listening environment.

In one embodiment, the invention operates to measure noise level picked up by one or more of the microphones in a multi-microphone directional processing system, and then either activating the directional processing when the noise level is high or deactivating the directional processing when the noise level is low. Additionally, transitions between activation and deactivation of the directional processing can be made smoothly without annoying users.

Consequently, the invention enables multi-microphone directional processing systems to achieve automatic directional processing when needed The invention is described below with respect to embodiments particularly well suited for use with hearing aid applications. However, it should be recognized that the invention is not limited to hearing aid applications, but is applicable to other sound pick-up systems.

Embodiments of the first aspect of the invention are discussed below with reference to FIGs. 2 - 8. However, those skilled in the art will readily appreciate that the detailed description given herein with respect to these figures is for explanatory purposes as the invention extends beyond these limited embodiments.

FIG. 2 is a block diagram of a two-microphone directional processing system 200 according to one embodiment of the invention. The two-microphone directional processing system 200 includes a first microphone 202 and a second microphone 204. The first microphone 202 produces a first electronic sound signal and the second microphone 204 produces a second electronic sound signal. A delay unit 206 delays the second electronic sound signal. The two-microphone directional processing system 200 also includes a minimum estimate unit 208 and an automatic on/off control unit 210. The minimum estimate unit 208 estimates a minimum level for the first electronic

sound signal. Typically, the minimum level is measured over a time constant duration, such that the minimum is a relatively long-term minimum. The automatic on/off control unit 210 produces a directional processing control signal that is sent to a multiplication unit 212. The multiplication unit 212 then multiplies the second electronic sound signal with the directional processing control signal at the multiplication unit 212 to produce a processed second electronic sound signal. The processed second electronic sound signal is thus processed to either perform directional processing or not perform directional processing. In one implementation, the multiplication unit 212 scales the second electronic sound signal by "1" when directional processing is to be performed, and scales the second electronic sound signal by "0" when directional processing is not to be performed. A subtraction unit 214 then subtracts the processed second electronic sound signal from the first electronic sound signal to produce an output signal. At this point, the output signal has undergone directional processing by the two-microphone directional processing system 200 when the noise level picked up by the first microphone 202 is sufficiently high. Such directional processing enables unwanted interference from certain directions to be suppressed. However, in cases where the noise level picked up by the first microphone 202 is low, the two-microphone directional processing system 200 does not perform directional processing. Consequently, the disadvantages of directional processing are automatically avoided when directional processing is not beneficial.

In this embodiment, minimum estimates and multiplication calculations are performed. The minimum estimates can, for example, be performed by minimum estimate units shown in more detail below with respect to FIGs. 3 and 4. It should also be noted that the delay unit 206 can be positioned within the two-microphone directional processing system 200 anywhere in the channel associated with the second electronic sound signal prior to the subtraction unit 214.

The minimum level being measured by the minimum estimate unit 208 represents an estimate of the noise level being packed-up by the first microphones. Although the two-microphone directional processing system 200 uses minimum estimates of the electronic sound signals produced by the first

and second microphones 202 and 204, other signal characteristics can alternatively be used to measure noise level. For example, Root-Mean-Square (RMS) average of the electronic sound signals produced by the microphones could be used. With such an approach, the RMS average could be measured over a time constant duration. The time constant can be set such that the average is relatively long-term so as to avoid impact of signal fluctuations. The time constant with an RMS approach is likely to be longer than the time constant for the minimum approach.

FIG. 3 is a block diagram of a minimum estimate unit 300 according to one embodiment of the invention. The minimum estimate unit 300 is, for example, suitable for use as the minimum estimate unit discussed above with respect to FIG. 2. The minimum estimate unit 300 receives an input signal (e.g., electronic sound signal) that is to have its minimum estimated. The input signal is supplied to an absolute value circuit 302 that determines the absolute value of the input signal. An add circuit 304 adds the absolute value of the input signal together with an offset amount 306 and thus produces an offset absolute value signal. The addition of the offset amount, which is typically a small positive value, such as 0.000000000001, is used to avoid overflow in division or logarithm calculations performed in subsequent circuitry in the multi-microphone directional processing systems. The offset absolute value signal from the add circuit 304 is supplied to a subtract circuit 308. The subtract circuit 308 subtracts a previous output 310 from the offset absolute value signal to produce a difference signal 312. The difference signal 312 is supplied to a multiply circuit 314. In addition, the difference signal 312 is supplied to a switch circuit 316. The switch circuit 316 selects one of two constants that are supplied to the multiply circuit 314. A first of the constants, referred to as alphaB, is supplied to the multiply circuit 314 when the difference signal 312 is greater than or equal to zero. Alternatively, a second constant, referred to as alphaA, is supplied to the multiply circuit 314 when the difference signal 312 is not greater than or equal to zero. The constants, alphaA and alphaB, are typically small positive values, with alphaA being greater than alphaB. In one implementation, alphaA is 0.00005 and alphaB is 0.000005. The multiply circuit 314 multiplies the difference signal 312 by the selected constant to

produce an adjustment amount. The adjustment amount is supplied to an add circuit 318. The add circuit 318 adds the adjustment amount to the previous output 310 to produce a minimum estimate for the input signal. A sample delay circuit 320 delays the minimum estimate by a delay (1/z) to yield the previous output 310 (where 1/z represents a delay operation).

FIG. 4 is a block diagram of a minimum estimate unit 400 according to another embodiment of the invention. The minimum estimate unit 400 is, for example, similar in design to the minimum estimate unit 300 illustrated in FIG. 3. The minimum estimate unit 400, however, further includes a linear-to-logarithm conversion unit 402 that converts the offset absolute value signal into a logarithmic offset signal before being supplied to the subtract circuit 308. The minimum estimate unit 400 is, for example, suitable for use as the minimum estimate unit discussed above with respect to FIG. 2. Optionally, a logarithm-to-linear conversion could be performed at the output of the minimum estimate circuit 400.

The two constants, alphaA and alphaB, are used in the minimum estimate units 300, 400 to determine how the minimum estimate changes with the input signal. Because the constant alphaA is greater than the constant alphaB, the minimum estimate tracks the valley level (or minimum level) of the input signal. Since the valley level is typically a good indicator of the noise level in the sound, the minimum estimate produced by the minimum estimate units 300, 400 is a good indicator of background noise level.

FIG. 5 is a block diagram of an automatic on/off control unit 500 according to one embodiment of the invention. The automatic on/off control unit 500 is, for example, suitable for use as the automatic on/off control unit 210 illustrated in FIG. 2. The automatic on/off control unit 500 includes a subtract circuit 502 and a subtract circuit 504. The subtract circuits 502 and 504 receive an input signal. The input signal, for example, represents the minimum estimate, such as the minimum estimate produced by the minimum estimate unit 208 illustrated in FIG. 2. The subtract circuit 502 also receives a second level setting (L2), and the subtract circuit 504 receives a first level setting (L1). The second level setting (L2) is greater than the first level setting (L1). The first level setting (L1) and the second level setting (L2) can be referred to as

threshold amounts, levels or values. The subtract circuit 502 subtracts the second level setting (L2) from the input signal to produce a first control signal that is supplied to a switch circuit 506. The subtract circuit 504 subtracts the input signal from the first level setting (L1) to produce a second control signal that is supplied to switch circuit 508. Note that the input signals to the automatic on/off control unit 500 pertains to a noise level picked-up by one or more of the microphones. When the first control signal indicates that the input signal (i.e., noise level) is greater than the second level setting (L2), the switch circuit 506 causes a constant "1" value to be supplied as an output of the automatic on/off control unit 500. Alternatively, when the switch circuit 508 determines that the second control signal is less than the first level setting (L1), the switch circuit 508 outputs a "0" value which is passed through the switch circuit 506 and thus forms the output. The output of the automatic on/off control unit 500 is also coupled to a sample delay circuit 510 that subjects the output signal to a delay on the order of one sample. In other words, the sample delay circuit 510 delays the output signal by a delay (1/z) to yield a previous output (or delayed output) (where 1/z represents a delay operation). The previous output is fed back as another input to the switch unit 508. Hence, when the input signal to the automatic on/off control unit 500 is between the first level setting (L1) and the second level setting (L2), the output signal is held in its previous state. In other words, the delayed output produced by the sample delay circuit 510 is passed back through the switch circuit 508 and then through the switch circuit 506 to again become the output.

FIG. 6 is a schematic diagram of an automatic on/off control unit 600 according to one embodiment of the invention. The automatic on/off control unit 600 is, for example, also suitable for use as the automatic on/off control unit 210 illustrated in FIG. 2. The automatic on/off control unit 600 includes a subtract circuit 602. The subtract circuit 602 receives an input signal to the automatic on/off control unit 600. The input signal represents the noise level picked up by one of the microphones, such as one of the microphones 202 and 204 illustrated in FIG. 2. The subtract circuit also receives a reference level (L). The reference level (L) can be referred to as a threshold amount, level or value. The subtract circuit 602 subtracts the reference level (L) from the input signal to

produce a value indicating the extent to which the input signal exceeds the reference level (L). This difference signal is then scaled by a scaling circuit 604. As an example, the scaling circuit can scale down the difference signal by 20% (0.05). The scaled difference signal produced by the scaling circuit 604 is then passed through a limit circuit 606 so that a resulting output signal has its amplitude limited to a value between 0 and 1.

FIG. 7 is a graph illustrating a relationship between directional processing (indicated by directional scale) and an input level for the automatic on/off control unit 500 illustrated in FIG. 5. FIG. 7 indicates that a smooth transition between directional processing "on" and directional processing "off" is achieved. In effect, the transitioning between directional processing "on" and directional processing "off" enjoys a hysteresis characteristic to prevent rapid oscillations in switching directional processing "on" and "off". More particularly, the first level setting (L1) is a constant that determines when to absolutely turn-off the directional processing, and the second level setting (L2) is a constant that determines when to absolutely turn-on the directional processing. When input signal (e.g., noise level) is less than the first level setting (L1), the directional scale is zero ("0") and directional processing is turned off. When the input is greater than the second level setting (L2), the directional scale is one ("1") and the directional processing is turned on. When the input is between the first level setting (L1) and the second level setting (L2), the directional scale does not change. That is, if the directional processing was "on" previously, it should stay "on". If the directional processing was "off" previously, it should stay "off". It is desirable to set the second level setting (L2) to be greater than the first level setting (L1). This is because the noise level usually does not vary much in a short time, thus setting the second level setting (L2) to be greater than the first level setting (L1) guarantees that the estimate variation of noise level by the "minimum estimate" will not frequently trigger the directional processing "on" and "off". Therefore, a smooth transition between the two stages is achieved.

FIG. 8 is a graph illustrating a relationship between directional processing (indicated by directional scale) and an input level for the automatic on/off control unit 600 illustrated in FIG. 6. When the input level is less than the

reference level (L) (threshold level), the directional processing is completely "off". As the input level exceeds the threshold level, the directional processing is gradually performed more and more as the input level increases up to the condition in which the directional processing is completely "on". More specifically, if the input signal (e.g., noise level) is less than the "threshold", the directional scale is zero and the directional processing is turned "off". If the input signal is greater than the "threshold", the directional scale gradually increases as the input level goes up. The rate of the increase is determined by the scaling rate of the scaling circuit 604. If directional scale is one, the directional processing is fully "on". If the directional scale is less than one but greater than zero, directional processing is on but less effective. Because the directional processing is gradually switched in as the noise level increases, the small variation in the noise estimate will not cause great change in the nature of the directional processing and therefore, the transition between directionality on and off is perceptually smooth.

The advantages of the first aspect of the invention are numerous. Different embodiments or implementations may yield one or more of the following advantages. One advantage of the invention is that directional processing to aid in interference suppression is automatically controlled. Another advantage of the invention is that directional processing is deactivated when not beneficial. Still another advantage of the invention is that directional processing is done in a manner that is perceptively smooth to the user.

According to a second aspect of the invention, the approaches adjust a delay adaptively so that a directional null is placed in the direction of a dominant noise source. This would produce maximum Signal-to-Noise Ratio (SNR) improvement across all conditions. In other words, the dominant noise source is attenuated (e.g., suppressed) but the desired sound from a particular direction is not attenuated.

Consequently, the invention enables multi-microphone directional processing systems to adaptively suppress a noise source. The invention is described below with respect to embodiments particularly well suited for use with hearing aid applications. However, it should be recognized that the

invention is not limited to hearing aid applications, but is applicable to other sound pick-up systems.

Embodiments of the second aspect of the invention are discussed below with reference to FIGs. 10 - 16. However, those skilled in the art will readily appreciate that the detailed description given herein with respect to these figures is for explanatory purposes as the invention extends beyond these limited embodiments.

FIG. 10 is a block diagram of a two-microphone directional processing system 1000 according to one embodiment of the invention. The two-microphone directional processing system 1000 includes a first microphone 1002 and a second microphone 1004. The first microphone 1002 produces a first electronic sound signal, and the second microphone 1004 produces a second electronic sound signal. The first and second electronic sound signals can be either analog or digital signals. In one implementation, the first and second microphones 1002 and 1004 are physically spaced by a distance of at least 3 mm. A delay unit 1006 delays the second electronic sound signal by a delay amount. A subtraction unit 1008 then subtracts the delayed second electronic sound signal from the first electronic sound signal to produce an output signal. At this point, the output signal has undergone directional processing by the two-microphone directional processing system 1000. Such directional processing enables unwanted interference from certain directions to be suppressed.

The two-microphone directional processing system 1000 also includes an optimal delay determination unit 1010. The output signal produced by the substation unit 1008 is supplied to the optimal delay determination unit 1010. The optimal delay determination unit 1010 determines a delay amount (e.g., optimal delay) that the delay unit 1006 should induce on the second electronic sound signal so that the directional null associated with the directional processing occurs at the direction of a noise source. The delay amount, or a corresponding control signal, is supplied to the delay unit 1006 where the delay is imposed. Hence, the optimal delay determination unit 1010 causes the delay amount for the delay unit 1006 to self-adjust based on the output energy (e.g., output signal) of the two-microphone directional processing system 1000. In

other words, the delay induced by the delay unit 1006 automatically adjusts based on the output energy.

When interfering noise is present, the total energy of the signals picked up by the microphones 1002 and 1004 are greater than the output energy would be if the interfering noise were not present. According to one embodiment, the delay amount for the delay unit 1006 can be adjusted so that the output of the two-microphone directional processing system 1000 has minimum energy. Because change in the delay amount does not change the system response to desired sound coming from 0 degrees, minimizing the output energy by adjusting the delay is equivalent to achieving a maximum attenuation of noise (assuming the desired sound is coming from 0 degrees).

The output signal of the directional processing system 1000 can be further processed by other processing functions. In the case of hearing aid applications, the output of the directional processing is further processed by other hearing aid functions such as amplification and noise suppression.

FIG. 11 shows a block diagram of an optimal delay determination unit 1100 according to one embodiment of the invention. The optimal delay determination unit 1100 is, for example, suitable for use as the optimal delay determination unit 1100. The optimal delay determination unit 1100 includes an energy estimator 1102 and a delay generator 1104. The energy estimator 1102 receives a feedback signal 1106. The feedback signal 1106 is the output signal produced by the directional processing system. The energy estimator 1102 receives the feedback signal 1106 and creates an energy signal 1108. The delay generator 1104 receives the energy signal 1108 and generates a delay signal 1110 (delay amount; control signal) based on the energy signal 1108. More particularly, the delay generator 1104 controls the delay amount induced by the delay unit 1006 in such a way that the output energy is statistically minimized and, therefore, the Signal-to-Noise Ratio (SNR) is maximized.

The energy estimator 1102 can create the energy signal 1108 by any one of the following: (1) forcing its input into positive signal; (2) squaring the input; (3) calculating a Root-Mean-Square (RMS) signal for the input; or (4)

estimating a minimum signal from the input. The energy signal 1108 can be down-sampled first before being used to generate the delay signal 1110.

The delay generator 1104 produces the delay signal 1110 based on the energy signal 1108. In one embodiment, the delay signal 1110 is a delay amount obtained by determining a change in the energy signal, creating a delay increment signal in accordance with the change, and adding the delay increment signal to a current delay amount to produce a next delay amount.

FIG. 12A is a block diagram of a delay generator 1200 according to one embodiment of the invention. The delay generator 1200 is, for example, suitable for use as the delay generator 1104 illustrated in FIG. 11. The delay generator 1200 includes a subtraction circuit 1202. The subtraction circuit 1202 receives the energy signal 1108 from the energy estimator 1102. A sample delay circuit 1204 delays the energy signal 1108 by a specified amount (e.g., 1/z) before supplying the delayed energy signal to the subtraction circuit 1202. The subtraction circuit 1202 subtracts the energy signal 1108 from the delayed energy signal to produce an energy change signal. The energy change signal is supplied to a delay increment calculation circuit 1206.

The delay increment calculation circuit 1206 calculates a current delay increment based on the energy change signal. The current delay increment is then supplied to an add circuit 1208. The add circuit 1208 adds the current delay increment to a previous delay increment 1209 to output an unrestricted optimal delay. The unrestricted optimal delay is then supplied to a maximum delay circuit 1210 and a minimum delay circuit 1212. The unrestricted optimal delay, after passing through the maximum delay circuit 1210 and the minimum delay circuit 1212, outputs an optimal delay 1216. The maximum delay circuit 1210 limits the upper range for the optimal delay to a maximum value, and the minimum delay circuit 1212 limits the minimum delay to a minimum value. Although the limits will vary widely with application, in one embodiment, the maximum value can be 36 and the minimum value can be zero. The optimal delay 1216 is also fed back through a sample delay circuit 1218 which produces the previous delay increment 1209 that is supplied to the add circuit 1208. The optimal delay 1216 is, for example, the delay signal 1110 illustrated in FIG. 11.

The circuitry for the delay increment calculation circuit 1206 can take many forms. FIGs. 12A, 12B and 12C illustrate three of many different approaches to calculate or determine the current delay increment.

FIG. 12B is a schematic diagram of a circuit 1220 suitable for use as the delay increment calculation circuit 1206 according to one embodiment of the invention. The circuit 1220 calculates the current delay increment from the energy change signal. The circuit 1220 includes a switch circuit 1222, a negate circuit 1224, and a sample delay circuit 1226. The energy change signal is supplied to a control terminal of the switch circuit 1222 to control its switching. The switch circuit 1222 outputs the delay increment signal. The delay increment signal is also fed back to the sample delay circuit 1226 which produces a previous delay increment signal. The previous delay increment signal is supplied to the negate circuit 1214 as well as to a first switch terminal of the switch circuit 1222. The negate circuit 1224 inverts the previous delay increment signal and supplies the inverted previous delay increment signal to a second switch terminal of the switch circuit 1222.

The switch circuit 1222 is controlled in accordance with the energy difference signal. When the switch circuit 1222 determines that the energy difference signal is greater than zero (0), then the delay increment signal being output by the circuit 1220 corresponds to the previous delay increment signal. Alternatively, when the switch circuit 1222 determines that the energy difference signal is less than zero (0), then the delay increment signal being output by the circuit 1220 corresponds to the inverted previous delay increment. Hence, when the energy difference signal is greater than zero (0), the delay increment signal remains the same as it previously was. On the other hand, when the energy difference signal is less than zero (0), then the delay increment signal is negated from its previous value. As an example, the energy difference signal and the delay increment being output can be represented in multiple bits, such as 16 bits, of either integer or floating point numerical storage.

FIG. 12C is a schematic diagram of a circuit 1240 suitable for use as the delay increment calculation circuit 1206 according to another embodiment of the invention. The circuit 1240 calculates the current delay increment from the

energy change signal. The circuit 1240 includes a multiply circuit 1242 and a sample delay circuit 1244. The energy difference signal is received at the multiply circuit 1242. In addition, the multiply circuit 1242 receives a previous delay increment signal from the sample delay circuit 1244. Here, the multiply circuit 1242 multiplies the energy difference signal with the previous delay increment signal to produce the delay increment signal. The delay increment signal is also supplied to the sample delay circuit 1244 which delays the signal by a specified amount (1/z) to produce the previous delay increment signal.

FIG. 12D is a schematic diagram of a circuit 1260 suitable for use as the delay increment calculation circuit 1206 according to still another embodiment of the invention. The circuit 1260 calculates the current delay increment from the energy change signal. The circuit 1260 includes a scaling circuit 1262, a multiply circuit 1264, and a sample delay circuit 1266. The energy difference signal is supplied to the scaling circuit 1262 that scales the energy difference signal in accordance with a parameter K. Here, in one embodiment, the scaling parameter K is negative (-K). The scaled energy difference signal is then supplied to the multiply circuit 1264. The multiply circuit 1264 also receives a previous delay increment signal produced by the sample delay circuit 1266. The multiply circuit 1264 multiplies the previous delay increment signal by the scaled energy difference signal to produce the delay increment signal. The delay increment signal is also supplied to the sample delay circuit 1266 which delays the signal by a specified amount (1/z) to produce the previous delay increment signal.

FIG. 13 is a block diagram of a two-microphone directional processing system 1300 according to another embodiment of the invention. The two-microphone directional processing system 1300 includes a first microphone 1302 and a second microphone 1304. The first microphone 1302 produces a first electronic sound signal, and the second microphone 1304 produces a second electronic sound signal. The first and second electronic sound signals can be either analog or digital signals.

The directional processing system 1300 also includes a series of different delay units 1306, 1308 and 1310. Each of these delay units 1306,

1308 and 1310 operate to induce different delays to the second electronic sound signal. In addition, the directional processing system 1300 also includes subtract circuits 1312, 1314 and 1316. Each of the subtract circuits 1312, 1314 and 1316 receives the first electronic sound signal from the first microphone 1304. In addition, the subtract circuit 1314 receives the delayed second electronic sound signal from the delay unit 1306. The subtract circuit 1314 receives the delayed second electronic sound signal from the delay unit 1308. The subtract circuit 1316 receives the delayed second electronic sound signal from the delay unit 1310. Each of the subtract circuits 1312, 1314 and 1316 produce a difference signal. The difference signals produced by the subtract circuits 1312, 1314 and 1316 are each supplied to a signal selection circuit 1318. Under the control of a control signal, the signal selection circuit 1318 outputs one of the difference signals as the output signal. At this point, the output signal has undergone directional processing by the directional processing system 1300. Such directional processing enables unwanted interference from certain directions to be suppressed.

The control signal to the signal selection circuit 1318 is provided by a selector 1320 together with energy estimators 1322, 1324 and 1326. The energy estimator 1322 estimates the energy on the difference signal produced by the subtract circuit 1312, and supplies the energy estimate to a first input to the selector 1320. The energy estimator 1324 estimates the energy on the difference signal produced by the subtract circuit 1314, and supplies the energy estimate as a second input to the selector 1320. The energy estimator 1326 estimates the energy of the difference signal produced by the subtract circuit 1316 and supplies the energy estimate to a third input to the selector 1320. The selector 1320 then selects one of the estimated energy values supplied by the energy estimators 1322, 1324 and 1326 as the selected output which forms the control signal that controls the signal selection circuit 1318.

The directional processing system 1300 selects the difference signal that has the lowest energy as the system output (output signal). The lowest energy means that the channel or path undergoing the most noise suppression is selected. The different delay units 1306, 1308 and 1310 together with the subtract units 1312, 1314 and 1316 for the channels or paths. In this

embodiment, the delays for the delay elements are fixed and thus do not adapt. Instead, the various delay units offer different delays and the channel or path providing the best noise suppression is chosen. Although the directional processing system 1300 provided only three channels or paths, it should be recognized that additional paths can be provided. In general, the directional processing system 1300 operates with two or some channels or paths.

The signal energy can be estimated in a variety of ways. For example, the energy signal can be estimated using one of the followings: (1) forcing its input into positive signal; (2) squaring the input; (3) calculating a Root-Mean-Square (RMS) signal for the input; or (4) estimating a minimum signal from the input. Also, it should be noted that the rate at which the energy signal is estimated need not be the same as the rate in which the delay signal is updated. In other words, the energy signal can be updated with a different time constant that a time constant used in updating the delay signal. For example, for a fixed sampling rate, the energy signal can be updated for every sample, while the delay signal can be updated every 100 samples.

The adaptive directional processing system includes at least two microphones, typically physically spaced by a distance of at least three (3) mm. The microphones are used to convert sound into electronic signals. The electronic signals can be either analog or digital. The system further includes delay means to delay the electronic signals from one or both microphones. The system further includes addition or subtraction means to generate a differential signal of the microphone outputs as delayed by the delay means. The system also includes means for estimating the energy of the differential signal. The delay from the delay means is used to adapt the directional null to suppress a dominant noise source. The delay means, the addition/subtraction means, and the energy estimate means can be used more than once in parallel so that multiple delayed signals, multiple differential signals, and multiple energy signals are created.

Although the above-described embodiments of the directional processing systems have utilized two microphones, it should be understood that the directional processing systems can also use more than two microphones. Furthermore, following directional processing, the output of the

directional processing system can be further processed by other processing functions. In the case of hearing aid applications, the output of the directional processing is further processed by other hearing aid functions such as amplification and noise suppression.

FIG. 14 is a graph illustrating a spectrum of a 1 kHz pure tone in white noise without any directional processing for noise reduction. The SNR of the spectrum is about 6 dB.

FIG. 15 is a graph illustrating a spectrum of a 1 kHz pure tone in white noise with fixed-pattern (hypercaidiod) directional processing for noise reduction. The SNR of the spectrum is about 14 dB.

FIG. 16 is a graph illustrating a spectrum of a 1 kHz pure tone in white noise with adaptive directional processing according to one embodiment of the invention to provide enhanced noise reduction. The SNR of the spectrum is about 30 dB, which is a dramatic improvement over the conventionally available SNRs associated with FIGs. 14 and 15.

The advantages of the second aspect of the invention are numerous. Different embodiments or implementations may yield one or more of the following advantages. One advantage of the invention is that a dominant noise source can be directionally suppressed. Another advantage of the invention is that the directional suppression is adaptive and thus changes as the directional of the dominant noise source changes. Still another advantage of the invention is that desired sound from a particular direction is not interfered with even though a dominant noise source is able to be directionally suppressed.

The invention can also be combined not only with the first and second aspects but also with other inventions so as to share circuitry or otherwise complement one another. For example, the invention described herein can be combined with the adaptive microphone sensitivity matching described in U.S. Application No. 09/_______, filed March 14, 2001, and entitled "ADAPTIVE MICROPHONE MATCHING IN MULTI-MICROPHONE DIRECTIONAL SYSTEM", the contents of which is hereby incorporated by reference.

The invention is preferably implemented in hardware, but can be implemented in software or a combination of hardware and software. The

invention can also be embodied as computer readable code on a computer readable medium. The computer readable medium is any data storage device that can store data which can be thereafter be read by a computer system. Examples of the computer readable medium include read-only memory, random-access memory, CD-ROMs, magnetic tape, optical data storage devices, carrier waves. The computer readable medium can also be distributed over a network coupled computer systems so that the computer readable code is stored and executed in a distributed fashion.

The many features and advantages of the present invention are apparent from the written description and, thus, it is intended by the appended claims to cover all such features and advantages of the invention. Further, since numerous modifications and changes will readily occur to those skilled in the art, it is not desired to limit the invention to the exact construction and operation as illustrated and described. Hence, all suitable modifications and equivalents may be resorted to as falling within the scope of the invention.

What is claimed is:

CLAIMS

1. A directional sound processing system, comprising:

at least first and second microphones spaced apart by a distance, said first microphone producing a first electronic sound signal and said second microphone producing a second electronic sound signal;

a noise level estimate circuit operatively coupled to said first or second microphone, said noise level estimate circuit operates to produce a noise level estimate associated with the first or second electronic sound signal from said first or second microphone; and

a directional processing circuit operatively connected to said first and second microphones and said noise level estimate circuit, said directional processing circuit operates to activated or deactivate directional processing with respect to the first and second electronic sound signals based on the noise level estimate.

2. A directional sound processing system as recited in claim 1, wherein when the noise level estimate is less than a threshold amount, said directional processing circuit deactivates the directional processing.

3. A directional sound processing system as recited in claim 1,

wherein when the noise level estimate is less than a first threshold amount, said directional processing circuit deactivates the directional processing, and

wherein when the noise level estimate is greater than a second threshold amount, said directional processing circuit activates the directional processing.

4. A directional sound processing system as recited in claim 3,

wherein the second threshold amount is greater than the first threshold amount, and

wherein when the noise level estimate is between the first threshold amount and the second threshold amount, said directional processing circuit does not change the activation or deactivation of the directional processing from its previous state.

5. A directional sound processing system as recited in claim 1, wherein said directional processing circuit comprises:

a directional processing control circuit operatively coupled to said noise level estimate circuit, said directional processing control circuit produces a control signal based on the noise level estimate and at least one threshold; and

a signal modification circuit operatively connected to said directional processing control circuit, said signal modification circuit operates to modify the second electronic sound signal in accordance with the control signal.

6. A directional sound processing system as recited in claim 5, wherein said directional processing circuit further comprises:

a combining circuit operatively connected to said signal modification circuit and said first microphone, said combining circuit operates to produce an output signal by combining the modified second electronic sound signal with the first electronic sound signal.

7. A directional sound processing system as recited in claim 6, wherein said directional sound processing system further comprises:

a delay circuit that delays the second electronic sound signal or the modified second electronic sound signal by a delay amount.

8. A directional sound processing system as recited in claim 6,

wherein the control signal is a scaling signal, and

wherein said signal modification circuit is a multiplication circuit that multiplies the second electronic sound signal with the control signal.

9. A directional sound processing system as recited in claim 6, wherein the control signal is one of a logical "1" and a logical "0".

10. A directional sound processing system as recited in claim 6, wherein said combining circuit is a subtraction circuit.

11. A directional sound processing system as recited in claim 1, wherein said directional sound processing system further comprises:

a delay circuit that delays the second electronic sound signal by a delay amount.

12. A directional sound processing system as recited in claim 1, wherein said directional processing circuit comprises:

a directional processing control circuit operatively coupled to said noise level estimate circuit, said directional processing control circuit operates to produce a control signal based on the noise level estimate and at least one threshold; and

a scaling circuit operatively connected to said directional processing control circuit, said scaling circuit operates to scale the second electronic sound signal in accordance with the control signal; and

a subtraction circuit operatively connected to said scaling circuit and said first microphone, said subtraction circuit operates to produce an output difference signal by subtracting the scaled second electronic sound signal from the first electronic sound signal.

13. A directional sound processing system as recited in claim 12, wherein said directional sound processing system further comprises:

a delay circuit that delays the second electronic sound signal or the scaled second electronic sound signal by a delay amount.

14. A directional sound processing system as recited in claim 1, wherein said directional sound processing system resides within a hearing aid device.

15. A directional sound processing system, comprising:

at least first and second microphones spaced apart by a distance, said first microphone producing a first electronic sound signal and said second microphone producing a second electronic sound signal;

a minimum estimate circuit operatively coupled to said first or second microphone, said minimum estimate circuit produces a minimum estimate for the first or second electronic sound signal from said first or second microphone;

a directional processing control circuit operatively coupled to said minimum estimate circuit, said directional processing control circuit produces a control signal based on the minimum estimate; and

a scaling circuit operatively connected to said directional processing control circuit, said scaling circuit operates to scale the second electronic sound signal in accordance with the control signal; and

a subtraction circuit operatively connected to said scaling circuit and said first microphone, said subtraction circuit producing an output difference signal by subtracting the scaled second electronic sound signal from the first electronic sound signal.

16. A directional sound processing system as recited in claim 15, wherein said directional sound processing system further comprises:

a delay circuit that delays the second electronic sound signal or the scaled second electronic sound signal by a delay amount.

17. A directional sound processing system as recited in claim 15, wherein said scaling circuit comprises a multiplier.

18. A directional sound processing system as recited in claim 15, wherein said directional sound processing system resides within a hearing aid device.

19. In a hearing aid device having a multi-microphone sound processing device, a method for dynamically controlling directional processing in the multi-microphone sound processing system, said method comprising:

(a) receiving first and second electronic sound signals from first and second microphones, respectively;

(b) producing a differential electronic sound signal based on the first and second sound signals when an estimated noise level is greater than a first threshold; and

(c) alternatively producing a non-differential sound signal based on the first and second sound signals when the estimated noise level is less than greater than a second threshold.

20. A method as recited in claim 19, wherein the first threshold is greater than or equal to the second threshold.

21. A method as recited in claim 19, wherein the first and second microphones are provided within a hearing aid device, and wherein said method is performed by the hearing aid device.

22. A method for dynamically controlling directional processing in the multi-microphone sound processing system, said method comprising:

(a) receiving first and second electronic sound signals from first and second microphones, respectively;

(b) estimating a noise level picked up by at least one of the first and second microphones; and

(c) dynamically controlling the directional processing based on the estimated noise level.

23. A method as recited in claim 22, wherein said controlling (c) comprises:

(c1) comparing the estimated noise level to at least one threshold level to produce a directional processing control signal; and

(c2) controlling the directional processing in accordance with the directional processing control signal.

24. A method as recited in claim 23, wherein said controlling (c2) comprises scaling one of the first and second electronic sound signals processing in accordance with the directional processing control signal.

25. A method as recited in claim 22, wherein said controlling (c) comprises:

(c1) comparing the estimated noise level to a threshold level to produce a comparison signal; and

(c2) deactivating the directional processing when the estimated noise level is below the threshold level.

26. A method as recited in claim 22, wherein said controlling (c) comprises:

(c1) comparing the estimated noise level to a first threshold level to produce a first comparison signal;

(c2) comparing the estimated noise level to a second threshold level to produce a second comparison signal, the second threshold level being greater than the first threshold level;

(c3) deactivating the directional processing when the estimated noise level is below the first threshold level; and

(c4) activating the directional processing when the estimated noise level is greater than the second threshold level.

27. A method as recited in claim 26, wherein the second threshold level is greater than the first threshold level.

28. A method as recited in claim 22, wherein the first and second microphones are provided within a hearing aid device, and wherein said method is performed by the hearing aid device.

29. A method as recited in claim 22, wherein the noise level is estimate by a minimum estimator.

30. An adaptive directional sound processing system, comprising:

a least two microphones spaced apart by a predetermined distance, each of said microphones producing an electronic sound signal;

a delay circuit that delays the electronic sound signal from at least one of said microphones by an adaptive delay amount;

a subtraction circuit operatively connected to said microphones and said delay circuit, said subtraction circuit producing an output difference signal from the electronic sound signals following said delay circuit; and

a delay amount determination circuit operatively coupled to receive the output difference signal, said delay amount determination circuit produces a

delay control signal that is supplied to said delay circuit so as to control the adaptive delay amount.

31. An adaptive directional sound processing system as recited in claim 30, wherein the adaptive delay amount varies so as to directionally suppress undesired sound.

32. An adaptive directional sound processing system as recited in claim 30, wherein the adaptive delay amount induced by said delay circuit operates to minimize the energy amount of the output difference signal.

33. An adaptive directional sound processing system as recited in claim 30, wherein the adaptive delay amount induced by said delay circuit operates to minimize the energy amount of the output difference signal while not significantly attenuating sound arriving at said microphones from a predetermined direction.

34. An adaptive directional sound processing system as recited in claim 30, wherein said adapting operates to minimize the energy amount of the output difference signal so as to maximize Signal-to-Noise Ratio (SNR).

35. An adaptive directional sound processing system as recited in claim 30, wherein said adaptive directional sound processing system resides within a hearing aid device.

36. An adaptive directional sound processing system, comprising:

a least two microphones spaced apart by a predetermined distance, each of said microphones producing an electronic sound signal;

a delay circuit that delays the electronic sound signal from at least one of said microphones by an adaptive delay amount;

a logic circuit operatively connected to said microphones and said delay circuit, said logic circuit producing an output signal from the electronic sound signals following said delay circuit; and

a delay amount determination circuit operatively coupled to receive the output signal, said delay amount determination circuit produces a delay control signal based on the output signal, the delay control signal being is supplied to said delay circuit so as to control the adaptive delay amount.

37. An adaptive directional sound processing system as recited in claim 36, wherein the adaptive delay amount varies so as to directionally suppress undesired sound.

38. An adaptive directional sound processing system as recited in claim 36, wherein the adaptive delay amount induced by said delay circuit operates to minimize the energy amount of the output signal.

39. An adaptive directional sound processing system as recited in claim 36, wherein the adaptive delay amount induced by said delay circuit operates to minimize the energy amount of the output signal while not significantly attenuating sound arriving at said microphones from a predetermined direction.

40. An adaptive directional sound processing system as recited in claim 36, wherein said adapting operates to minimize the energy amount of the output signal so as to maximize Signal-to-Noise Ratio (SNR).

41. An adaptive directional sound processing system as recited in claim 36, wherein said adaptive directional sound processing system resides within a hearing aid device.

42. An adaptive directional sound processing system as recited in claim 36, wherein the adaptive delay amount induced by said delay circuit is controlled such that a delay increment is added to a previously determined adaptive delay amount.

43. An adaptive directional sound processing system as recited in claim 42, wherein the delay increment is determined based on change in energy on the output signal.

44. An adaptive directional sound processing system as recited in claim 42, wherein the change in energy selects one of two possible delay increments.

45. An adaptive directional sound processing system as recited in claim 44, wherein the two possible delay increments are a previous delay increment and an inverse previous delay increment.

46. An adaptive directional sound processing system as recited in claim 42, wherein the delay increment is determined by multiplying a previous delay increment by a change in energy on the output signal.

47. An adaptive directional sound processing system as recited in claim 42, wherein the delay increment is determined by scaling a change in energy on the output signal and then multiplying a previous delay increment by the change in energy on the output signal.

48. An adaptive directional sound processing system, comprising:

at least two microphones spaced apart by a predetermined distance, each of said microphones producing an electronic sound signal;

a delay circuit that delays the electronic sound signal from at least one of said microphones by an adaptive delay amount;

logic means for producing an output signal from the electronic sound signals following said delay circuit; and

delay determination means for producing a delay control signal based on the output signal, the delay control signal being is supplied to said delay circuit so as to control the adaptive delay amount.

49. A method for adaptively controlling delay induced on a sound signal so that unwanted noise is directionally suppressed, said method comprising:

(a) producing a difference signal from at least first and second sound signals respectively obtained by first and second microphones;

(b) estimating an energy amount of the difference signal; and

(c) producing a delay signal to control a delay amount induced on at least one of the first and second sound signals based on the energy amount of the difference signal.

50. A method as recited in claim 49, wherein said method further comprises:

(d) inducing the delay amount on at least one of the first and second sound signals.

51. A method as recited in claim 50, wherein following said inducing (d) said method (e) repeats said operations (a) – (d) so that the delay amount is dynamically adjusted so as to directionally suppress the unwanted noise.

52. A method as recited in claim 49, wherein the sound signal is provided by a hearing aid, and wherein said method is performed by the hearing aid.

53. An adaptive delay method for directional noise suppression in a hearing aid device, the hearing aid device having at least first and second microphones, said method comprising:

receiving first and second microphone outputs;

delaying at least the second microphone output by an adaptive delay amount;

combining the first microphone output and the delayed second microphone output to produce an output signal;

estimating an energy amount associated with the output signal; and

adapting the adaptive delay amount based on the energy amount.

54. A method as recited in claim 53, wherein said adapting operates to minimize the energy amount of the output signal while not significantly attenuating sound arriving at the first and second microphones from a predetermined direction.

55. A method as recited in claim 53, wherein said adapting operates to minimize the energy amount of the output signal so as to maximize Signal-to-Noise Ratio (SNR).

56. A method as recited in claim 53, wherein said combining comprises adding the first microphone output and the delayed second microphone output.

57. A method as recited in claim 53, wherein said combining comprises subtracting the first microphone output and the delayed second microphone output.

58. A method as recited in claim 53, wherein said adapting determines the adaptive delay amount based on change in energy on the output signal.

59. A method as recited in claim 58, wherein the change in energy on the output signal selects one of two possible delay increments.

60. A method as recited in claim 59, wherein the two possible delay increments are a previous delay increment and an inverse previous delay increment.

61. A method as recited in claim 53, wherein said adapting of the adaptive delay amount comprises multiplying a previous delay increment by a change in energy on the output signal.

62. A method as recited in claim 53, wherein said adapting of the adaptive delay amount comprises scaling a change in energy on the output signal and then multiplying a previous delay increment by the change in energy on the output signal.

63. A method for adaptively controlling delay induced on a sound signal in a multi-microphone directional processing system so that unwanted noise is directionally suppressed, said method comprising:

(a) receiving at least first and second sound signals respectively obtained by first and second microphones;

(b) delaying at least one of the first and second sound signals by a plurality of different delay amounts;

(c) producing, following said delaying (b), a plurality difference signals from at least first and second sound signals respectively obtained by first and second microphones;

(d) estimating energy amounts for each of the difference signals; and

(e) choosing the one of the difference signals as an output of the directional processing system based on the energy amounts of the difference signals.

64. A method as recited in claim 63, wherein the sound signals are provided by a hearing aid, and wherein said method is performed by the hearing aid.

65. An adaptive directional sound processing system, comprising:

at least two microphones spaced apart by a predetermined distance, each of said microphones producing an electronic sound signal;

a plurality of delay circuits that each delay the electronic sound signal from at least one of said microphones by a different delay amount;

logic means for producing candidate output signals from the electronic sound signals following said delay circuits; and

output selection means for selecting one of the candidate output signals as an output based on energy levels of the candidate output signals.

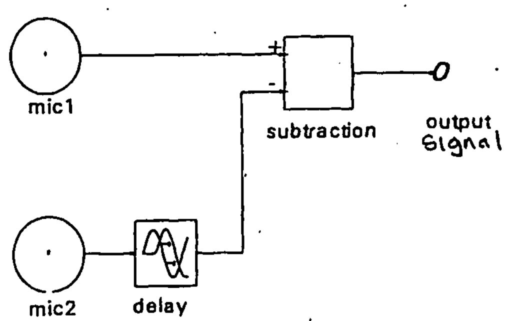

+
-
mic1
subtraction
output
Signal
mic2
delay

FIG. 1

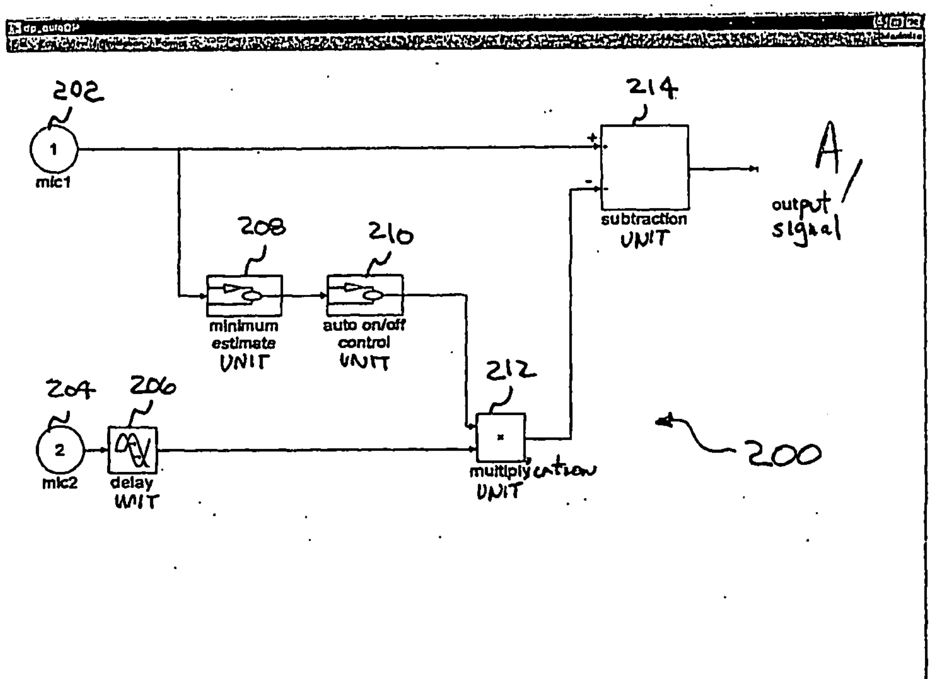
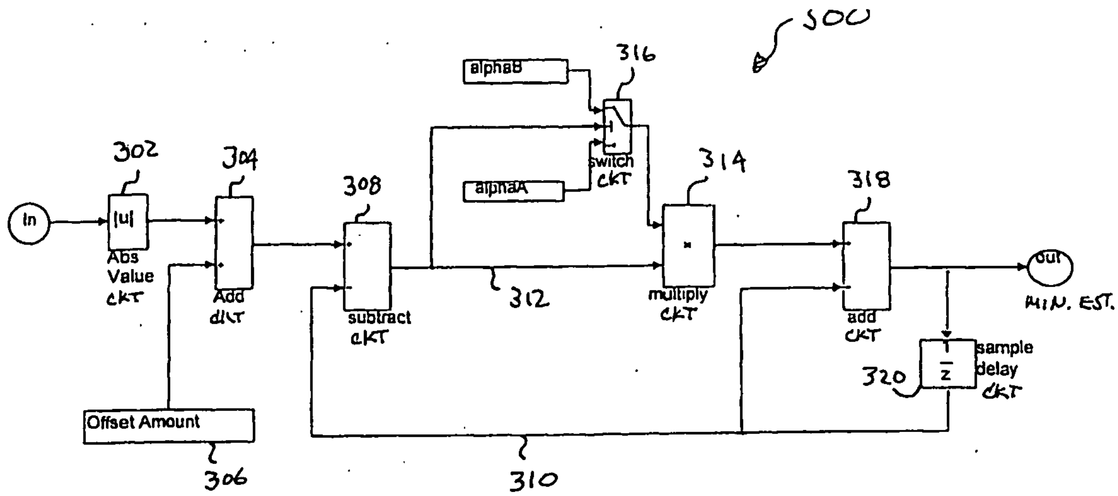

202
1
mic1
214
+
-
subtraction
UNIT
A
output
signal
208
minimum
estimate
UNIT
210
auto on/off
control
UNIT
212
multiplycation
UNIT
204
2
mic2
206
delay
UNIT
200

FIG. 2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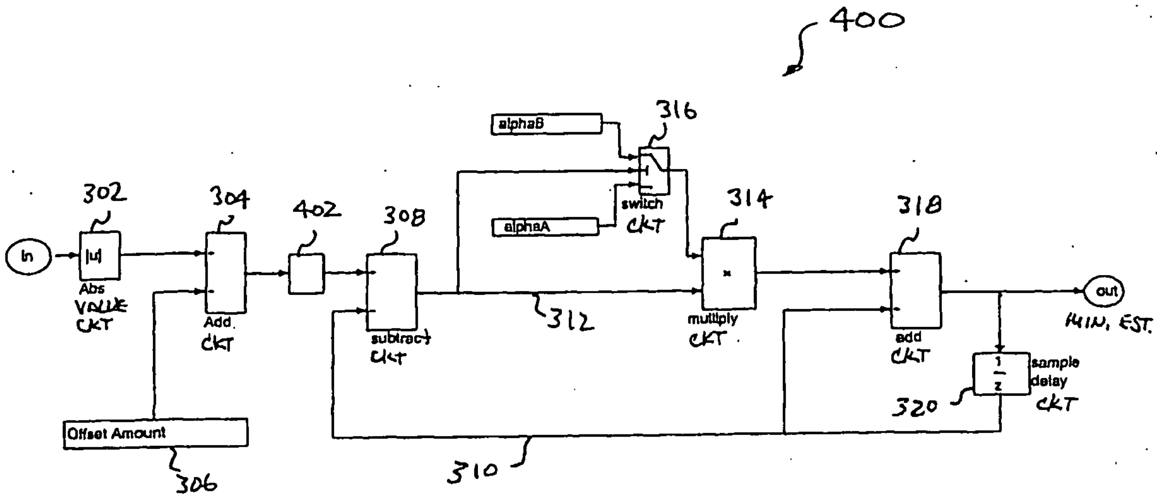
alphaB
316
alphaA
switch CKT
302
304
308
314
318
In
|u|
Abs Value CKT
Add CKT
subtract CKT
312
multiply CKT
add CKT
Out
MIN. EST.
sample delay CKT
320
Offset Amount
306
310

FIG. 3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alphaB
316
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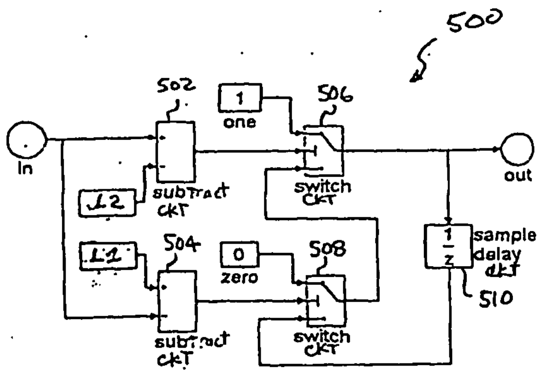
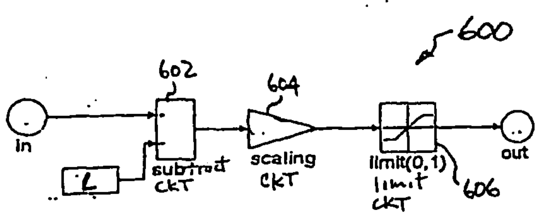
CKT
alphaA
302
|u|
Abs
VALUE
CKT
304
Add
CKT
402
308
subtract
CKT
312
314
multiply
CKT
318
add
CKT
out
MIN. EST.
In
1/z
sample
delay
CKT
320
Offset Amount
306
310

FIG. 4



L2 > L1
If in >= L2, out = 1
If in <= L1, out = 0
If L1<in<L2, out = previous out

FIG. 5



out = (in - threshold)*scaling
0 <=out <= 1

FIG. 6

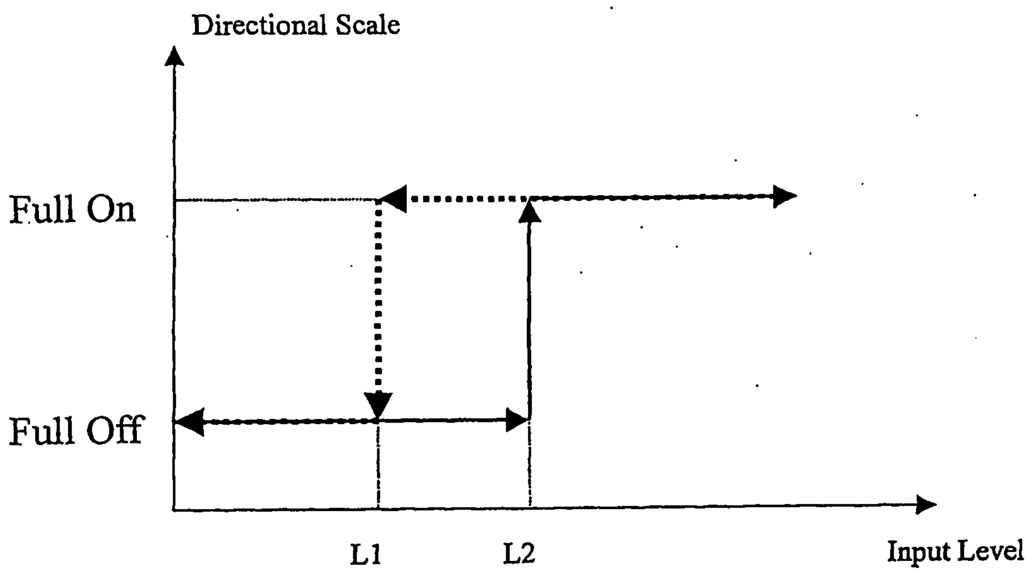


FIG. 7

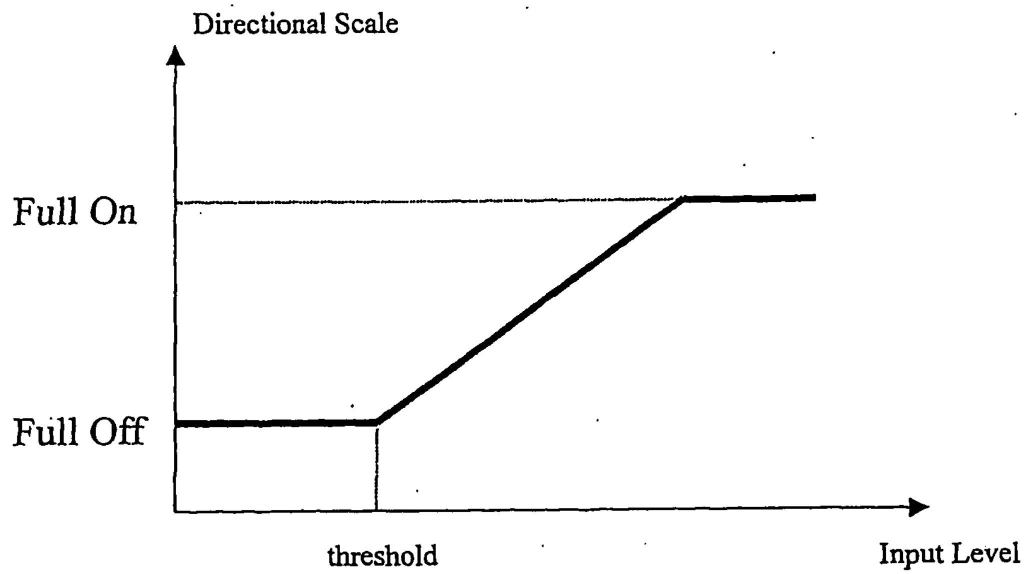
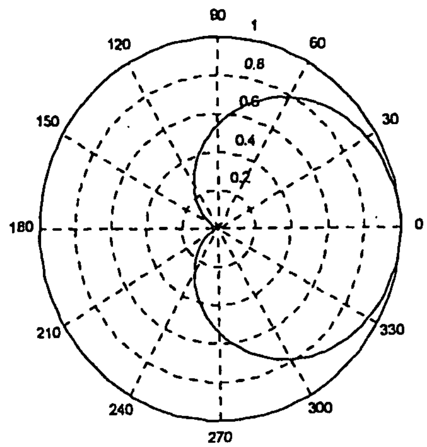
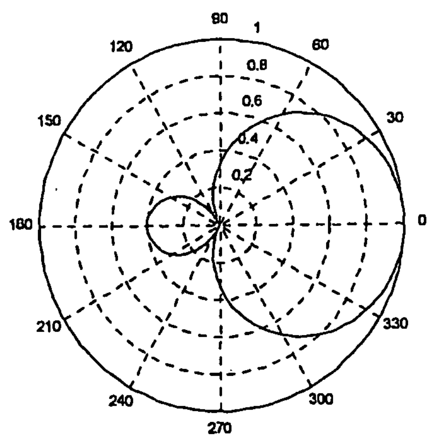


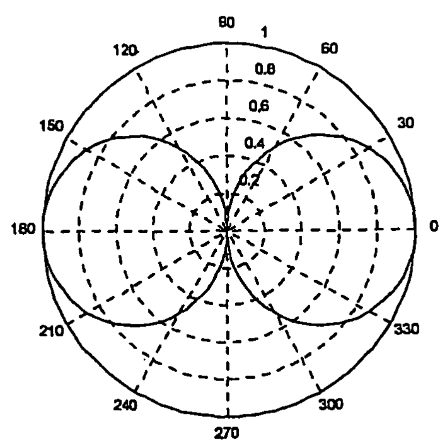
FIG. 8



9(a)



9(b)



9(c)

FIGs 9(a) -9(c)

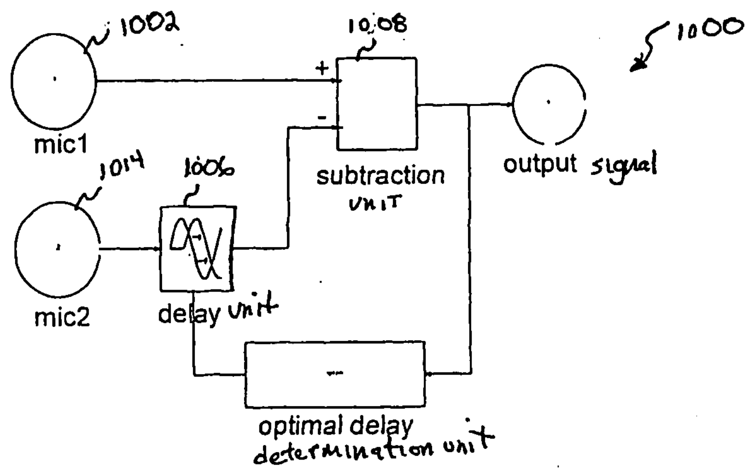

1002
1008
1000
+
-
mic1
1014
1006
subtraction unit
output signal
mic2
delay unit
optimal delay determination unit

FIG. 10

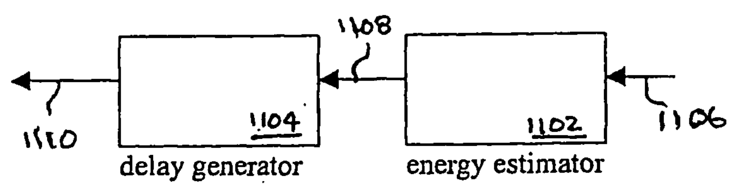

1108
1104
1102
1106
1110
delay generator
energy estimator

FIG. 11

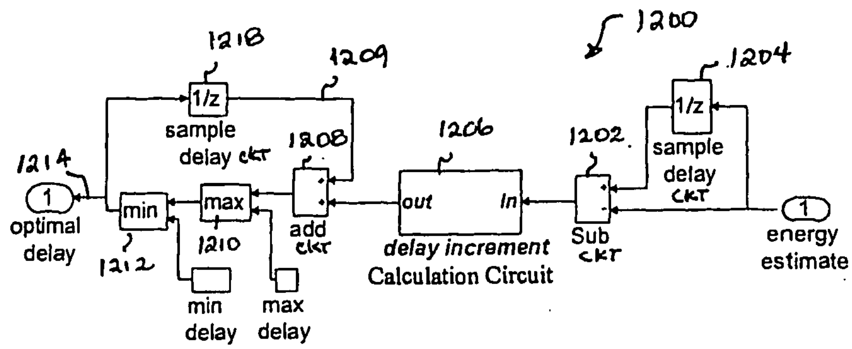


FIG. 12A

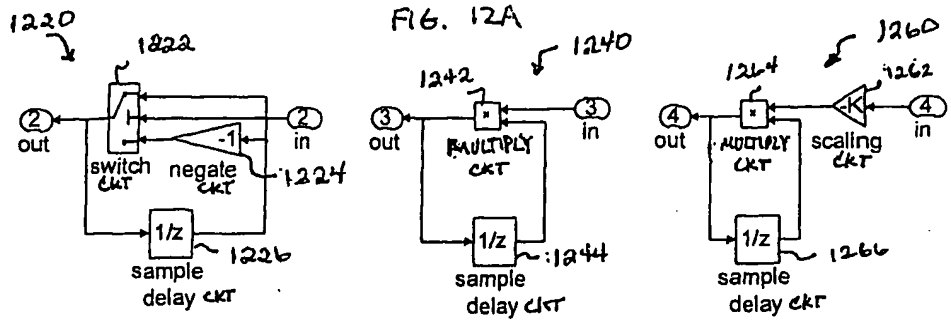


FIG. 12B FIG. 12C FIG. 12D

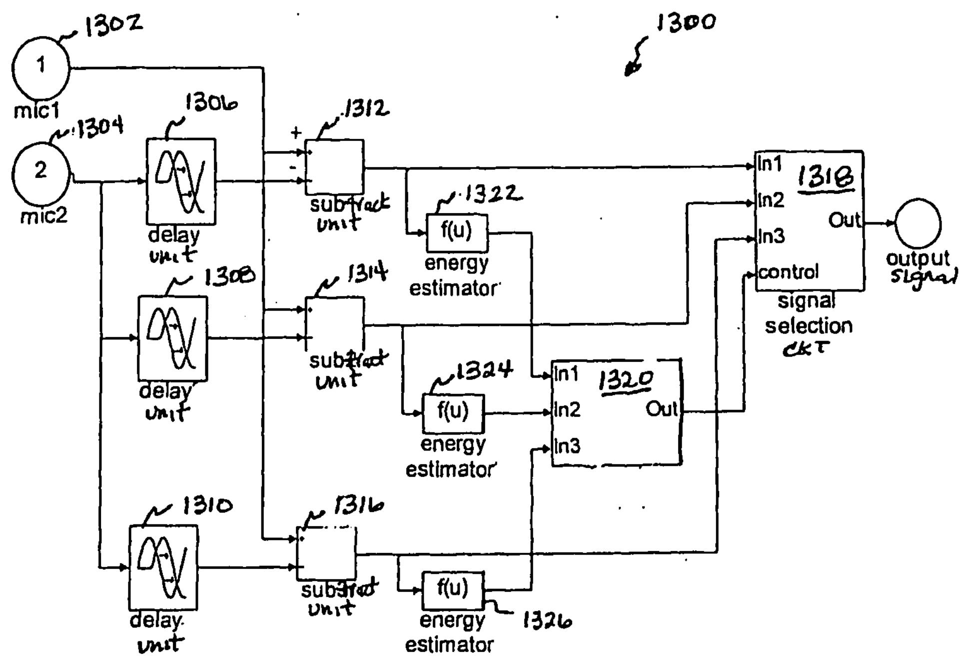

1300
1302
1
mic1
1304
2
mic2
1306
delay unit
1308
delay unit
1310
delay unit
1312
+
-
subtract unit
1314
subtract unit
1316
subtract unit
1322
f(u)
energy estimator
1324
f(u)
energy estimator
1326
f(u)
energy estimator
1320
In1
In2
In3
Out
1318
In1
In2
In3
control
Out
signal selection ckt
output signal

FIG. 13

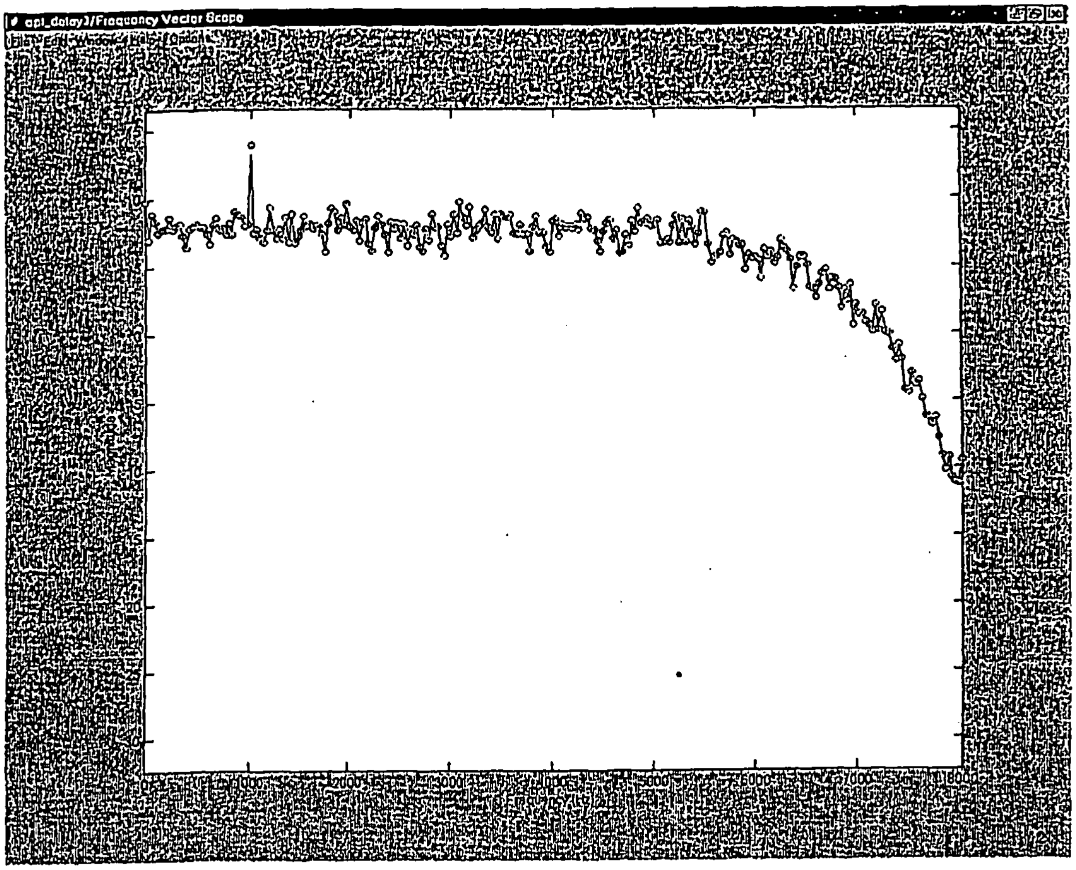

opt_delay3/Frequency Vector Scope
File Edit Window Help Options

FIG. 14

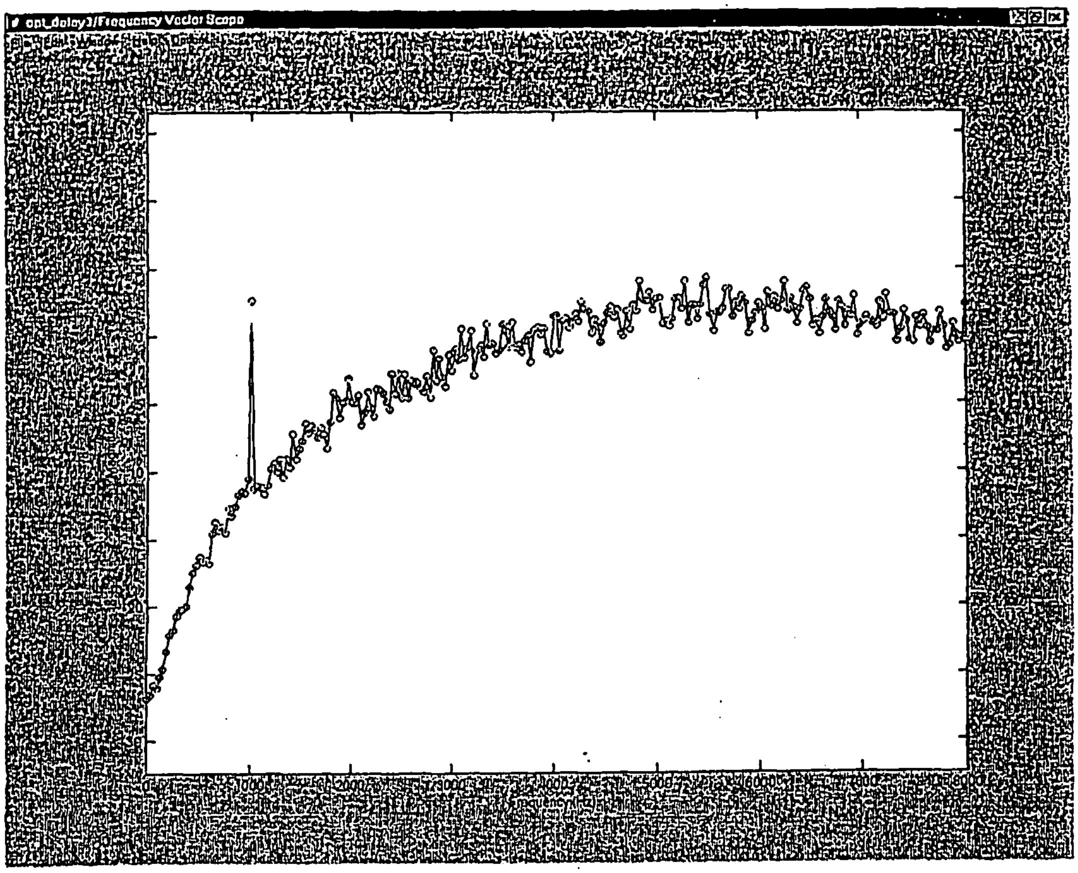


FIG. 15

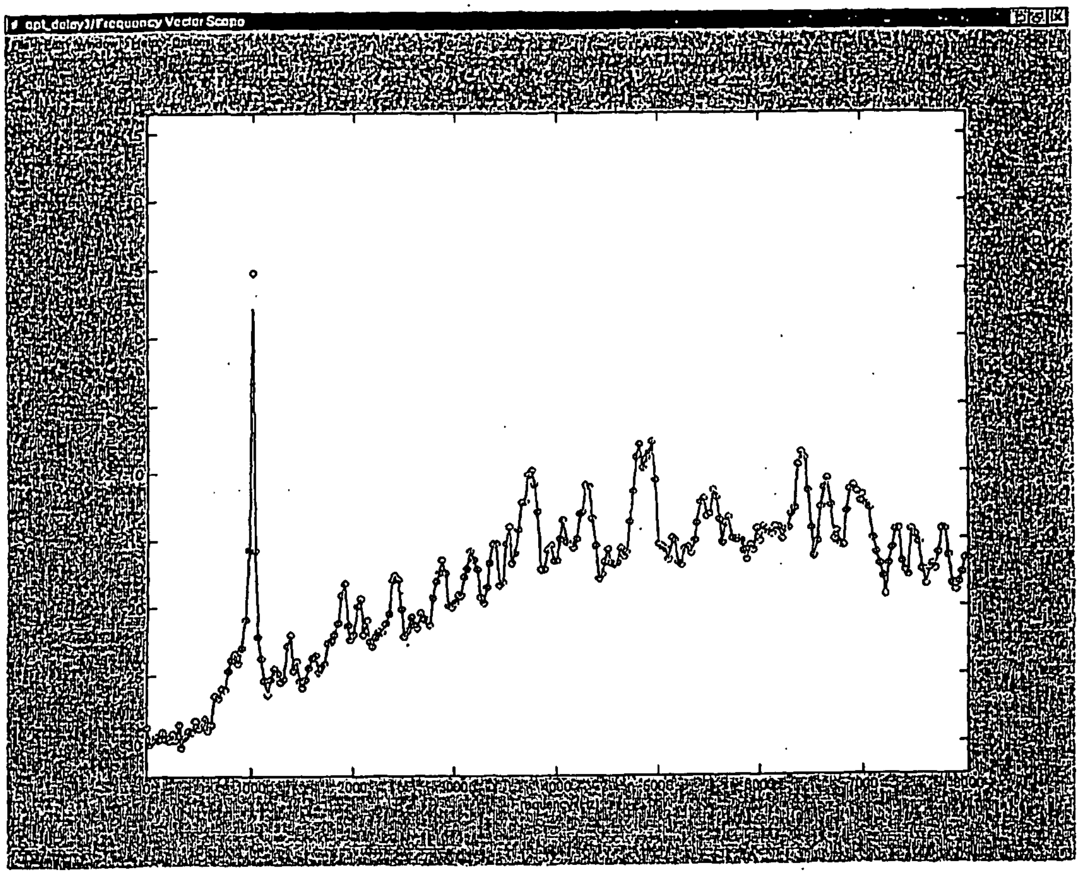


FIG. 16

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